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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/869,605	07/16/2001	Kim V. Hansen	66722-010-7	5723

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EXAMINER	
GRAHAM, ANDREW R	

ART UNIT	PAPER NUMBER
2644	7

DATE MAILED: 08/02/2004

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)
	09/869,605	HANSEN, KIM V.
	Examiner	Art Unit
	Andrew Graham	2644

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☐ Responsive to communication(s) filed on _____.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-6 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-6 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on 16 July 2001 is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☒ Some * c)☐ None of:

1.☒ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date 6.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

Information Disclosure Statement

1. The information disclosure statement filed September 13, 2001 was filed after the mailing date of the national stage application. In view of the data provided in the specification, the submission is in compliance with the provisions of 37 CFR 1.97. Accordingly, the information disclosure statement has been considered by the examiner.

Priority

2. Acknowledgment is made of applicant's claim for foreign priority based on an application filed in July 16, 2001. It is noted, however, that applicant has not filed a certified copy of the foreign application (EP 99610002) as required by PCT Rule 17.

Claim Rejections - 35 USC § 112

The following is a quotation of the second paragraph of 35 U.S.C. 112:

> The specification shall conclude with one or more claims particularly pointing out and distinctly claiming the subject matter which the applicant regards as his invention.

3. **Claims 1-4 and 6** are rejected under 35 U.S.C. 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention.

Claim 1 includes the limitation "mutually different directionality". The word "mutually" is considered indefinite herein because it is unclear what scope of interpretation is intended to be attributed to this word beyond that which is associated with "different". In other words, if A is "different" from B, then by definition, B is "different" from A, and the two are thus considered "mutually different". The use of such a word may be intended to convey the concept that the sound fields received by the two microphones are mutually exclusive or non-overlapping, but the current wording of the claim does not distinctly claim as such. Appropriate correction or clarification is required, noting the scope of definition afforded by the currently submitted disclosure.

Claims 2-4 are rejected due to their respective dependencies upon Claim 1.

Claim 6 also includes the phrase "mutually different" in the eighth line of the claim, and is considered indefinite for the same reason listed above in regards to Claim 1.

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

4. **Claims 1-4 and 6** are rejected under 35 U.S.C. 103(a) as being unpatentable over Jourjine et al (USPN 6526148 B1) in view of Bell (USPN 5706402) and Killion et al (USPN 5524056). Hereafter, "Jourjine et al" and "Killion et al" will be referred to as "Jourjine" and "Killion".

Jourjine teaches a method of blind source selection (BSS) for use with a hearing aid that involves a channel selection scheme. The teachings particularly apply to the improved pickup of a desired input signal with the attenuation of competing or undesired signals (col. 3, lines 6-16). The overall device thereby equates to, "A device for use in reducing noise in an audio signal containing noise and a target signal". Figure 1 illustrates that the input to the system is received through a pair of microphones (102,104), which reads on "at least two input channels each receiving a signal from a respective microphone" (col. 3, lines 17-35). The input from the microphones (102,104) is received by a signal processing device (200), which includes a calibration unit (204), an estimation/demixing unit (206), and a channel selection unit (208) (col. 3, lines 31-44). These components collectively read on "signal processing means in connection with the input channels". A speaker (202) is provided with the output of the signal processor (200), for emitting the processed inputs as audible sound (col. 3, lines 32-36). This component equates to "a receiver in connection with the signal processing means". The processing involved uses a fast BSS technique, involving the selection of one of the microphone input signals based on the best signal to

noise ratio (SNR), and therefore best intelligibility, of the input signals (col. 3, lines 53-67 and col. 4, lines 1-3). The involvement of the SNRs of the inputs equates to "based on differences of signal-to-noise ratios of the input signal in relation to a desired target signal". The attenuation of competing or background sounds equates to "removing at least part of the unwanted signal elements, thereby enhancing other parts of the audio signal". Section A of Jourjine provides a more detailed explanation of the derivation and calculation of the parameters involved with such delay and attenuation of the input signals (col. 4-7). Regarding the BSS technique, Jourjine teaches that the channel selection centers around the formulation of a mixing matrix and not a specific used BSS technique. Jourjine also provides an example that uses second order statistics, but notes that higher order methods may be employed (col. 7, lines 33-40).

Accordingly, Jourjine does not clearly specify:

- the processing by means of an independent component analysis

However, Bell teaches a blind signal processing system that involves unsupervised learning rules that solve the blind signal-processing problem (col. 9, lines 18-25). Such a system involves the reduction of redundancy in the involved neural network through the maximization of the mutual information, which then results in the minimization of relationships between separate outputs (col. 14, lines 53-65). The process detailed by Bell is denominated as "independent component analysis", which reads on "process the signals by means of an independent component analysis method". The process, again,

involves the maximizing the joint entropy between input signals, which reads on "determining whether statistical dependant signal elements are present" (col. 15, lines 15-35).

To one of ordinary skill in the art at the time the invention was made, it would have been obvious to utilize the teachings of Bell as part of the BSS solving method of Jourjine. The motivation behind such a modification would have been that the teachings of Bell minimize the statistical dependence of the two output signals.

However, Jourjine in view of Bell does not specify:

- that the two microphones of the system have mutually different directionality

Killion discloses a hearing aid with at least two microphones, and a processing scheme that reduces the SNR of the produced output signal (col. 6, lines 21-43). One embodiment involves the use of an omnidirectional microphone (15) and a directional microphone (20), while another embodiment involves the use of two directional microphones (col. 5, lines 15-21 and col. 6, lines 37-48). Both systems involve a switch (55) which alternately connects the microphone signals to the output of the system (col. 6, lines 22-30 and 63-64). Figures 18 and 19 illustrate the sound ports (400,400',415,415',440) associated with a pair of first order directional microphones (445,450) and an omnidirectional microphone (col. 11, lines 35). Figure 19 particularly shows diffration scoops (480) for increasing effective spacing and directional sensitivity of the microphones (col. 11, lines 46-56). Collectively, these teachings

and the combination of various directional and non-directional microphones, reads on "at least two of the microphones having mutually different directionality".

To one of ordinary skill in the art at the time the invention was made, it would have been obvious to include microphones of different directionalities as taught by Killion in the hearing aid system of Jourjine in view of Bell. The motivation behind such a modification would have been that such microphones would have enabled different pickup sound fields to be provided as inputs to the processing system, which would have improved the SNR of an input in regards to an input source with a predetermined directionality.

Regarding **Claim 2,** the embodiment of Figure 13 includes an omnidirectional microphone (230), a first order directional microphone (235), and a second order directional microphone (240) (col. 9, lines 46-49). This reads on "the device comprises at least a directional microphone and an omnidirectional microphone".

Regarding **Claim 3,** Figure 1 illustrates a SDPT switch (55) which exclusively connects either of the two microphones (15,20) to the hearing aid amplifier (60) (col. 6, lines 22-35). The embodiment of Figure 13 illustrates the use of FET switches (255,260,275) connected to amplifiers (280,285) such that different levels of noise conditions enable the inputs of different microphones to dominate the output signal (col. 9, lines 51-67). These two switching arrangements read on "means are provided for switching between the two or more output signals or combinations of these".

Regarding **Claim 4**, the embodiment of Figure 13 illustrates the use of FET switches (255,260,275) connected to amplifiers (280,285) such that different levels of noise conditions enable the inputs of different microphones to dominate the output signal (col. 9, lines 51-67). The operation of the amplifiers (280,285) and switches (255,260,275) is based directly on a rectified (270) copy of the input signal from the omnidirectional microphone (230) (col. 9, lines 54-56). This reads on "two or more output signals are produced and where automatic switching means are provided for switching between the two or more output signals according to a predetermined scheme".

Regarding **Claim 6**, please refer above to the components cited in the rejection of Claim 1, noting particularly the amplifier (60) and microphones with different directionalities of the teachings of Killion.

5. **Claims 1-4 and 6** are rejected under 35 U.S.C. 103(a) as being unpatentable over Strandberg (USPN 6023514) in view of Kellermann (USPN 5602962) and Killion et al (USPN 5524056). Hereafter, "Killion et al" will be referred to as "Killion".

Strandberg discloses a system for isolating the independent components of a merged audio field. Hearing aids are disclosed as one such system wherein such isolation of different audio sources is desirable (col. 1, lines 27-52 and col. 4, lines 55-61). The system is discussed in terms of a target source (16b) in the context of two other sources (16a,16c) (col. 5, lines 30-41). One or more of the

factored signals (38a-38c), which correspond to the signals (14a-14c) from the sources (16a-16c), may be selectively transmitted to a user (col. 7, lines 44-49). This selective further processing, in the context of a target source, reads on "A device for use in reducing noise in an audio signal containing noise and a target signal" and "removing at least part of the unwanted signal elements, thereby enhancing other parts of the audio signal". The exemplary embodiment of Strandberg shows three sensors, but it is noted that two may be used (col. 5, lines 1-2 and 58-67 and col. 6, lines 1-2). This reads on "at least two channels each receiving a signal from a respective microphone". The input signals are then digitized and applied to a digital signal processor (col. 6, lines 24-51). This equates to "signal processing means in connection with the input channels". In the context of a hearing aid device, the "transmitted to the user" would inherently involve some form of acoustic output device, which equates to "a receiver in connection with the digital processing means" (col. 7, lines 47-49). The individual components the form the merged acoustic field are based on cross-correlation, wherein the maximum element in the cross-correlation array is used to calculate the original source signal (col. 9, lines 7-58). This processing is executed by the digital signal processor (20). This isolation of independent components based on finding a maximum cross-correlation reads on "adapted to process the signals by means of an independent component analysis" and "determining whether statistical dependent signal elements are present".

However, Strandberg does not specify:

- that the processing involves differences of signal to noise ratios of the inputs signals in relation to a desired target signal

Kellermann discloses a system for processing an input comprising both speech and noise components. Input for the system is provided through the use of N microphones, which are connected to N inputs of a preprocessor unit (2) (col. 3, lines 3-19). The signals are also applied to controllable multipliers (3), wherein the control signal for the multipliers is obtained from an evaluation unit (4) in response to the input signals (col. 3, lines 25-31). The control signals are based on the ratio of the speech component in each signal to the power of the noise component, which corresponds to the signal to noise ratio (col. 2, lines 1-3 and col. 4, lines 10-49). The produced effect of such processing is an improved overall signal to noise ratio of the output signal and thereby an improved speech audibility (col. 5, lines 24-29). This formation of a control signal or weighting factor, reads on "based on differences of signal-to-noise ratios of the input signals in relation to a desired target signal".

To one of ordinary skill in the art at the time the invention was made, it would have been obvious to include the input weighting scheme of Kellerman as part of the signal processing in the system of Strandberg. The motivation behind such a modification would have been the improvement in the suppression of a noise or undesired component of a received audio signal. The scheme of Kellermann also

incorporates the efficient computation of weighting factors in real-time, which would have been desirable for implementation in an alternate system, such as a hearing aid.

However, Strandberg in view of Kellermann does not specify:

- that the two microphones of the system have mutually different directionality

Killion discloses a hearing aid with at least two microphones, and a processing scheme that reduces the SNR of the produced output signal (col. 6, lines 21-43). One embodiment involves the use of an omnidirectional microphone (15) and a directional microphone (20), while another embodiment involves the use of two directional microphones (col. 5, lines 15-21 and col. 6, lines 37-48). Both systems involve a switch (55) which alternately connects the microphone signals to the output of the system (col. 6, lines 22-30 and 63-64). Figures 18 and 19 illustrate the sound ports (400,400′,415,415′,440) associated with a pair of first order directional microphones (445,450) and an omnidirectional microphone (col. 11, lines 35). Figure 19 particularly shows diffration scoops (480) for increasing effective spacing and directional sensitivity of the microphones (col. 11, lines 46-56). Collectively, these teachings and the combination of various directional and non-directional microphones, reads on "at least two of the microphones having mutually different directionality". The system of Killion is also noted to include a speaker, which also reads on "a receiver in connection with the signal processing means" (col. 11, lines 13-16).

To one of ordinary skill in the art at the time the invention was made, it would have been obvious to include microphones of different directionalities as taught by Killion in the hearing aid system of Strandberg in view of Kellermann. The motivation behind such a modification would have been that such microphones would have enabled different pickup sound fields to be provided as inputs to the processing system, which would have improved the SNR ratio of an input signal in regards to an input source with a predetermined directionality. Strandberg notes that microphones with such different directionalities have been known to be used in noise canceling systems.

Regarding **Claim 2,** the embodiment of Figure 13 includes an omnidirectional microphone (230), a first order directional microphone (235), and a second order directional microphone (240) (col. 9, lines 46-49). This reads on "the device comprises at least a directional microphone and an omnidirectional microphone".

Regarding **Claim 3,** Figure 1 illustrates a SDPT switch (55) which exclusively connects either of the two microphones (15,20) to the hearing aid amplifier (60) (col. 6, lines 22-35). The embodiment of Figure 13 illustrates the use of FET switches (255,260,275) connected to amplifiers (280,285) such that different levels of noise conditions enable the inputs of different microphones to dominate the output signal (col. 9, lines 51-67). These two switching arrangements read on "means are provided for switching between the two or more output signals or combinations of these".

Regarding **Claim 4**, the embodiment of Figure 13 illustrates the use of FET switches (255,260,275) connected to amplifiers (280,285) such that different levels of noise conditions enable the inputs of different microphones to dominate the output signal (col. 9, lines 51-67). The operation of the amplifiers (280,285) and switches (255,260,275) is based directly on a rectified (270) copy of the input signal from the omnidirectional microphone (230) (col. 9, lines 54-56). This reads on "two or more output signals are produced and where automatic switching means are provided for switching between the two or more output signals according to a predetermined scheme".

Regarding **Claim 6**, please refer above to the components cited in the rejection of Claim 1, noting particularly the amplifier (60) and microphones with different directionalities of the teachings of Killion.

6. **Claim 5** is rejected under 35 U.S.C. 103(a) as being unpatentable over Jourjine in view of Bell and Preves et al (USPN 5757933). Hereafter, "Preves et al" will be referred to as "Preves".

Jourjine teaches a method of blind source selection (BSS) for use with a hearing aid that involves a channel selection scheme. The teachings particularly apply to the improved pickup of a desired input signal with the attenuation of competing or undesired signals (col. 3, lines 6-16). The overall device thereby equates to, "A device for use in reducing noise in an audio signal containing noise and a target signal". Figure 1 illustrates that the input to the system is

received through a pair of microphones (102,104), which reads on "at least two input channels each receiving a signal from a respective microphone" (col. 3, lines 17-35). The input from the microphones (102,104) is received by a signal processing device (200), which includes a calibration unit (204), an estimation/demixing unit (206), and a channel selection unit (208) (col. 3, lines 31-44). These components collectively read on "signal processing means in connection with the input channels". A speaker (202) is provided with the output of the signal processor (200), for emitting the processed inputs as audible sound (col. 3, lines 32-36). This component equates to "a receiver in connection with the signal processing means". The processing involved uses a fast BSS technique, involving the selection of one of the microphone input signals based on the best signal to noise ratio (SNR), and therefore best intelligibility, of the input signals (col. 3, lines 53-67 and col. 4, lines 1-3). The involvement of the SNRs of the inputs equates to "based on differences of signal-to-noise ratios of the input signal in relation to a desired target signal". The attenuation of competing or background sounds equates to "removing at least part of the unwanted signal elements, thereby enhancing other parts of the audio signal". Section A of Jourjine provides a more detailed explanation of the derivation and calculation of the parameters involved with such delay and attenuation of the input signals (col. 4-7). Regarding the BSS technique, Jourjine teaches that the channel selection centers around the formulation of a mixing matrix and not a specific used BSS technique. Jourjine also

provides an example that uses second order statistics, but notes that higher order methods may be employed (col. 7, lines 33-40).

Accordingly, Jourjine does not clearly specify:

- the processing by means of an independent component analysis

However, Bell teaches a blind signal processing system that involves unsupervised learning rules that solve the blind signal-processing problem (col. 9, lines 18-25). Such a system involves the reduction of redundancy in the involved neural network through the maximization of the mutual information, which then results in the minimization of relationships between separate outputs (col. 14, lines 53-65). The process detailed by Bell is denominated as "independent component analysis", which reads on "process the signals by means of an independent component analysis method". The process, again, involves the maximizing the joint entropy between input signals, which reads on "determining whether statistical dependant signal elements are present" (col. 15, lines 15-35).

To one of ordinary skill in the art at the time the invention was made, it would have been obvious to utilize the teachings of Bell as part of the BSS solving method of Jourjine. The motivation behind such a modification would have been that the teachings of Bell minimize the statistical dependence of the two output signals.

However, Jourjine in view of Bell does not specify:

- that the at least two microphone means include beamforming means to make beamforming of the input signals from the

microphones and hereby adding directionality to the signal of the signal of at lease one of at least two input channels

Preves discloses a system that utilizes two microphones (B,F) and various components to shape the received sound field of the combined microphones (col. 4, lines 48-55). Switching means (S1) determine if the processed signal includes the non-directional input from just one of these microphones, or a processed, directional combination of these two microphones (col. 4, lines 61-67 and col. 5, lines 1-3). The second microphone (B) signal is passed through an inverter (52), an adjustable phase delay (54), and an adjustable gain (56), such that the user may vary the polar directivity pattern of the pickup of the hearing aid (col. 5, lines 14-26). The switch and these processing components read on "beamforming means to make a beamforming of the input signals from the microphones and hereby adding directionality to the signal of at least one of at least two input channels".

To one of ordinary skill in the art at the time the invention was made, it would have been obvious to include the switch and other components of the system of Preves as part of the microphone pickup of the system of Jourjine in view of Bell. The motivation behind such a modification would have been that such circuitry would have enabled a polar directivity pattern of the input to be varied from non-directional to super cardioid.

7. **Claim 5** is rejected under 35 U.S.C. 103(a) as being unpatentable over Strandberg in view of Kellermann and Preves et al (USPN 5757933). Hereafter, "Preves et al" will be referred to as "Preves".

Strandberg discloses a system for isolating the independent components of a merged audio field. Hearing aids are disclosed as one such system wherein such isolation of different audio sources is desirable (col. 1, lines 27-52 and col. 4, lines 55-61). The system is discussed in terms of a target source (16b) in the context of two other sources (16a,16c) (col. 5, lines 30-41). One or more of the factored signals (38a-38c), which correspond to the signals (14a-14c) from the sources (16a-16c), may be selectively transmitted to a user (col. 7, lines 44-49). This selective further processing, in the context of a target source, reads on "A device for use in reducing noise in an audio signal containing noise and a target signal" and "removing at least part of the unwanted signal elements, thereby enhancing other parts of the audio signal". The exemplary embodiment of Strandberg shows three sensors, but it is noted that two may be used (col. 5, lines 1-2 and 58-67 and col. 6, lines 1-2). This reads on "at least two microphones". The input signals are then digitized and applied to a digital signal processor (col. 6, lines 24-51). This equates to "signal processing means in connection with the input channels". In the context of a hearing aid device, the "transmitted to the user" would inherently involve some form of acoustic output device, which equates to "a receiver in connection with the digital

processing means" (col. 7, lines 47-49). The individual components the form the merged acoustic field are based on cross-correlation, wherein the maximum element in the cross-correlation array is used to calculate the original source signal (col. 9, lines 7-58). This processing is executed by the digital signal processor (20). This isolation of independent components based on finding a maximum cross-correlation reads on "adapted to process the signals by means of an independent component analysis" and "determining whether statistical dependent signal elements are present".

However, Strandberg does not specify:

- that the processing involves differences of signal to noise ratios of the inputs signals in relation to a desired target signal

Kellermann discloses a system for processing an input comprising both speech and noise components. Input for the system is provided through the use of N microphones, which are connected to N inputs of a preprocessor unit (2) (col. 3, lines 3-19). The signals are also applied to controllable multipliers (3), wherein the control signal for the multipliers is obtained from an evaluation unit (4) in response to the input signals (col. 3, lines 25-31). The control signals are based on the ratio of the speech component in each signal to the power of the noise component, which corresponds to the signal to noise ratios (col. 2, lines 1-3 and col. 4, lines 10-49). The produced effect of such processing is an improved overall signal to noise ratio of the output signal and thereby an improved speech

audibility (col. 5, lines 24-29). This formation of a control signal or weighting factor, reads on "based on differences of signal-to-noise ratios of the input signals in relation to a desired target signal".

To one of ordinary skill in the art at the time the invention was made, it would have been obvious to include the input weighting scheme of Kellermann as part of the signal processing in the system of Strandberg. The motivation behind such a modification would have been the improvement in the suppression of a noise or undesired component of a received audio signal. The scheme of Kellermann also incorporates the efficient computation of weighting factors in real-time, which would have been desirable for implementation in an alternate system, such as a hearing aid.

However, Strandberg in view of Kellermann does not specify:

- that the at least two microphone means include beamforming means to make beamforming of the input signals from the microphones and hereby adding directionality to the signal of the signal of at lease one of at least two input channels

Preves discloses a system that utilizes two microphones (B,F) and various components to shape the received sound field of the combined microphones (col. 4, lines 48-55). Switching means (S1) determine if the processed signal includes the non-directional input from just one of these microphones, or a processed, directional combination of these two microphones (col. 4, lines 61-67 and col. 5, lines 1-3). The second microphone (B) signal is passed through an inverter (52), an adjustable phase delay (54), and an adjustable gain (56), such that

the user may vary the polar directivity pattern of the pickup of the hearing aid (col. 5, lines 14-26). The switch and these processing components read on "beamforming means to make a beamforming of the input signals from the microphones and hereby adding directionality to the signal of at least one of at least two input channels".

To one of ordinary skill in the art at the time the invention was made, it would have been obvious to include the switch and other components of the system of Preves as part of the microphone pickup of the system of Strandberg in view of Kellermann. The motivation behind such a modification would have been that such circuitry would have enabled a polar directivity pattern of the input to be varied from non-directional to super cardioid.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Andrew Graham whose telephone number is 703-308-6729. The examiner can normally be reached on Monday-Friday, 8:30 AM to 5:00 PM (EST).

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Bill Isen can be reached on (703)305-4386. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

AG
Andrew Graham
Examiner
A.U. 2644

ag
July 26, 2004

XU MEI
PRIMARY EXAMINER